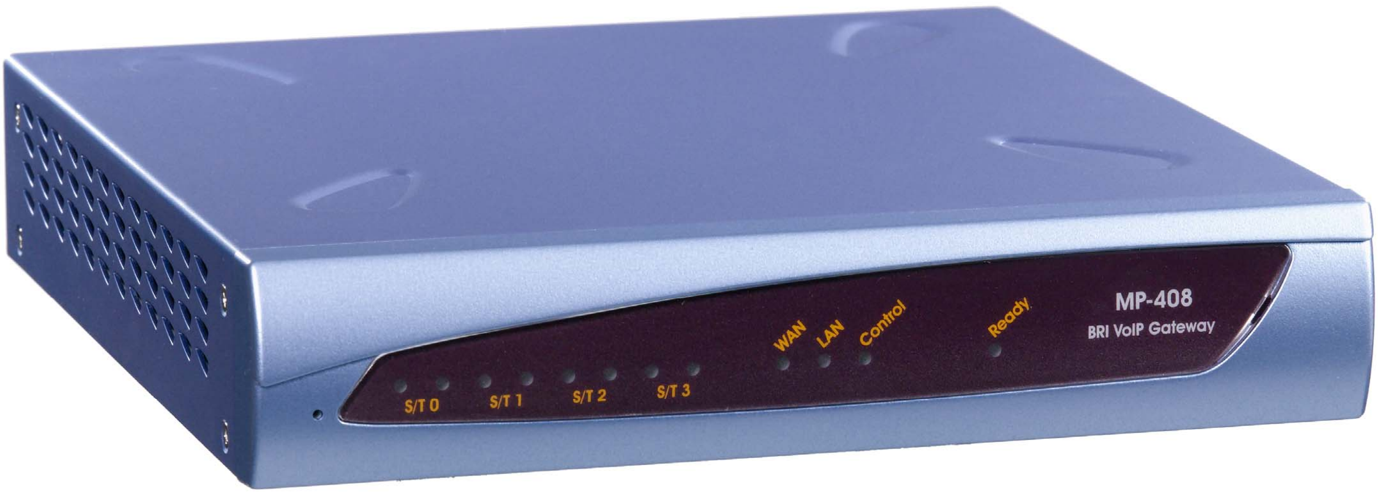
AudioCodes™ ISDN Media Gateways & CPE

SIP MediaPack™ MP-40x

Release Notes

Beta Version 2.0.7.5476



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Notice

This document describes the release of the AudioCodes MediaPack MP-40x BRI series Voice-over-IP (VoIP) media gateways.

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Each abbreviation, unless widely used, is spelled out in full when first used, and only Industry standard terms are used throughout this manual. The symbol 0x indicates hexadecimal notation.

Related Documentation

Document #	Manual Name
LTRT-837xx (e.g., LTRT-83701)	MP-40x SIP User's Manual
LTRT-839xx	MP-40x Case Reporting Templates

Note: MP-40x refers to the MP-408, MP-404, and MP-402 ISDN VoIP gateways.

Note: These Release Notes describe the MP-408, MP-404, and MP-402 ISDN VoIP gateways. Unless otherwise specified, whenever reference is made to the MediaPack in these Release Notes, it automatically includes these gateways.

1 What's New in Release Beta 2.0.7.5476

1.1 Supported Hardware Platforms

1.1.1 New Products Introduced in this Release

Model	Description
MP-402 /BRI /ST /AC /LL	MediaPack 402 ISDN VoIP gateway with single BRI interface (2 voice channels), LAN and WAN 10/100BaseT, AC power supply
MP-404 /BRI /ST /AC /FB	MediaPack 404 ISDN VoIP gateway with dual BRI interface (4 voice channels), with fallback configuration option, LAN and WAN 10/100BaseT, AC power supply
MP-404 /BRI /ST /AC /LL	MediaPack 404 ISDN VoIP gateway with dual BRI interface (4 voice channels), with lifeline support, LAN and WAN 10/100BaseT, AC power supply
MP-408 /BRI /ST /AC /FB	MediaPack 408 ISDN VoIP gateway with quad BRI interface (8 voice channels), with fallback configuration option, LAN and WAN 10/100BaseT, AC power supply
MP-408 /BRI /ST /AC /LL	MediaPack 408 ISDN VoIP gateway with quad BRI interface (8 voice channels), with lifeline support, LAN and WAN 10/100BaseT, AC power supply

1.2 Resolved Constraints

1.2.1 SIP

1. Interoperability problems with eyeBeam's SIP soft phone has been resolved.
2. SDP version is now correctly handled.

1.2.2 ISDN

1. Rejection of the Layer 3 Suspend Request has been resolved. Consequently, pressing PARK on the phone doesn't cancel the call.
2. The digit collection problem on ISDN user side has now being resolved.

1.2.3 SIP / ISDN Gateway

1. Call resources are no longer lost on the ISDN Network side.

1.2.4 IP

1. Fixed `ping` and `traceroute`.

1.2.5 Web Management

1. All parameters are now configurable by Web management.
2. All parameters are now checked upon entry and submitting.
3. All parameters and some table headers now provide a Help text.
4. Increased internal locking now prevents crashes during Web management.

1.2.6 CLI Management

1. CLI obsolete commands and error messages were removed.
2. The CLI prefix 'pstn' was replaced with 'isdn' for port, interface, and debug configuration commands.
3. The CLI prefix 'bluebox' was replaced with 'manipulation' for routing configuration and general setup commands.

1.2.7 Maintenance

1. The Reset button (located on the MP-40x device) is now functioning correctly.
2. No more crashes occur during SW upgrade using the Web management.

1.3 New and Modified Parameters

1. The default LAN static IP address was changed to 192.168.2.1.
2. In the ‘SIP Users’ screen, a new parameter ‘Authorization User’ for authentication was added.

2 SIP Compatibility

2.1 Supported SIP Features

The gateway supports the following SIP features:

- SIP using UDP transport layer.
- Works with Proxy or without Proxy, using an internal routing table.
- Fallback to internal routing table if Proxy is not responding.
- Supports up to four Proxy servers. If the primary Proxy fails, the MediaPack automatically switches to a redundant Proxy.
- Supports domain name resolving using DNS records for Proxy, Registrar and domain names that appear in the Contact and Record-Route headers.
- Proxy or Registrar Registration (per gateway or per gateway endpoint). The REGISTER message is sent to the Registrar's IP address (if configured) or to the Proxy's IP address. The message is sent per gateway or per configured user according to the "Registration Mode" parameter.
- Proxy and Registrar Authentication (handling 401 and 407 responses) using Basic or Digest methods. Accepted challenges are kept for future requests to reduce the network traffic.
- Configuration of authentication username and password per each gateway endpoint, or single username and password per gateway.
- Supported methods: INVITE, CANCEL, BYE, ACK, REGISTER, REFER and PRACK.
- Modifying connection parameters for an already established call (re-INVITE).
- Working with Redirect server and handling 3xx responses.
- Early media (supporting 183 Session Progress).
- PRACK reliable provisional responses (RFC 3262).
- Call Hold and Transfer Supplementary services using REFER, Refer-To, Referred-By, Replaces and NOTIFY.
- Call Forward (using 3xx response): Immediate, Busy, No reply, Busy or No reply, Do Not Disturb.
- Supports RFC 3581, Symmetric Response Routing.
- Supports Session Timers in SIP.
- Supports network asserted identity (RFC 3325).
- RFC 2833 Relay for DTMF Digits, including payload type negotiation.
- Supports RFC 2833, DTMF relay
- SIP URL: sip:"phone number"@IP address (such as 122@10.1.2.4, where "122" is the phone number of the source or destination phone number) or sip:"phone_number"@"domain name", such as 122@myproxy.com. Note that the SIP URI host name can be configured differently per called number.
- Negotiates coder from a list of given coders.

- Supported coders: G.711 A-law, G.711 μ-law, G.723.1, G.726 and G.729.
- Supports negotiation of dynamic payload types

2.1.1 Unsupported SIP Features

The following SIP features are NOT supported:

- MESSAGE, OPTION and NOTIFY method
- Preconditions (RFC 3312)
- SDP - Simple Capability Declaration (RFC 3407)
- Proxy discovery using NAPTR DNS records
- GRUU
- SIP over TCP

2.2 SIP Compliance Tables

The MediaPack gateways comply with RFC 3261, as shown in the following sections.

2.2.1 SIP Functions

Table 2-1: SIP Functions

Function	Supported
User Agent Client (UAC)	Yes
User Agent Server (UAS)	Yes
Proxy Server	Third-party only (tested with Ubiquity, Delta3, Microsoft, 3Com, BroadSoft, Snom and Cisco Proxies)
Redirect Server	Third-party
Registrar Server	Third-party

2.2.2 SIP Methods

Table 2-2: SIP Methods

Method	Supported	Comments
INVITE	Yes	
ACK	Yes	
BYE	Yes	
CANCEL	Yes	
REGISTER	Yes	Send only
REFER	Yes	
NOTIFY	Yes	
INFO	No	
OPTIONS	Yes	Used for keep alive
PRACK	Yes	

2.2.3 SIP Headers

The following SIP Headers are supported by the SIP gateway:

Table 2-3: SIP Headers (continues on pages 11 to 12)

Header Field	Supported
Accept	No
Accept–Encoding	No
Alert-Info	No
Allow	Yes
Also	No
Asserted-Identity	Yes
Authorization	Yes
Call-ID	Yes
Call-Info	No
Contact	Yes
Content-Disposition	No
Content-Encoding	Yes
Content-Length	Yes
Content-Type	Yes
Cseq	Yes
Diversion	No
Encryption	No
Expires	Yes
Fax	No
From	Yes
History-Info	No
Join	No
Max-Forwards	Yes
Messages-Waiting	No
MIN-SE	Yes
Organization	No
P-Asserted-Identity	Yes
P-Preferred-Identity	No
Priority	No
Proxy- Authenticate	Yes
Proxy- Authorization	Yes
Proxy- Require	Yes
Prack	Yes
Reason	No
Record- Route	Yes
Refer-To	Yes
Referred-By	Yes
Replaces	Yes
Require	Yes
Remote-Party-ID	No
Response- Key	No
Retry- After	No

Table 2-3: SIP Headers (continues on pages 11 to 12)

Header Field	Supported
Route	No
Rseq	Yes
Session-Expires	Yes
Server	No
SIP-If-Match	No
Subject	No
Supported	Yes
Timestamp	No
To	Yes
Unsupported	No
User- Agent	No
Via	Yes
Voicemail	No
Warning	No
WWW- Authenticate	Yes

2.2.4 SDP Headers

The following SDP Headers are supported by the SIP gateway:

Table 2-4: SDP Headers

SDP Header Element	Supported
v - Protocol version	Yes
o - Owner/ creator and session identifier	Yes
a - Attribute information	Yes
c - Connection information	Yes
d - Digit	No
m - Media name and transport address	Yes
s - Session information	Yes
t - Time alive header	No
b - Bandwidth header	No
u - Uri Description Header	No
e - Email Address header	No
i - Session Info Header	No
p - Phone number header	No
y - Year	No

2.2.5 SIP Responses

The following SIP responses are supported by the SIP gateway:

- 1xx Response - Information Responses.
- 2xx Response - Successful Responses.
- 3xx Response - Redirection Responses.
- 4xx Response - Client Failure Responses.
- 5xx Response - Server Failure Responses.
- 6xx Response - Global Responses.

2.2.5.1 1xx Response – Information Responses

Table 2-5: 1xx SIP Responses

1xx Response		Supported	Comments
100	Trying	Yes	The SIP gateway generates this response upon receiving of Proceeding message from ISDN or immediately after placing a call for CAS signaling.
180	Ringing	Yes	The SIP gateway generates this response for an incoming INVITE message. On receiving this response, the gateway waits for a 200 OK response.
181	Call is being forwarded	Yes	The SIP gateway does not generate these responses. However, the gateway does receive them. The gateway processes these responses the same way that it processes the 100 Trying response.
182	Queued	Yes	The SIP gateway generates this response in Call Waiting service. When SIP gateway receives 182 response, it plays a special waiting Ringback tone to TEL side.
183	Session Progress	Yes	The SIP gateway generates this response if Early Media feature is enabled and if the gateway plays a Ringback tone to IP

2.2.5.2 2xx Response – Successful Responses

Table 2-6: 2xx SIP Responses

2xx Response		Supported	Comments
200	OK	Yes	
202	Accepted	Yes	

2.2.5.3 3xx Response – Redirection Responses

Table 2-7: 3xx SIP Responses

3xx Response		Supported	Comments
300	Multiple Choice	Yes	The gateway responds with an ACK and resends the request to first in the contact list, new address.
301	Moved Permanently	Yes	The gateway responds with an ACK and resends the request to new address.
302	Moved Temporarily	Yes	The SIP gateway generates this response when call forward is used, to redirect the call to another destination. If such response is received, the calling gateway initiates an INVITE message to the new destination.
305	Use Proxy	Yes	The gateway responds with an ACK and resends the request to new address.
380	Alternate Service	Yes	The gateway responds with an ACK and resends the request to new address.

2.2.5.4 4xx Response – Client Failure Responses

Table 2-8: 4xx SIP Responses (continues on pages 14 to 15)

4xx Response		Supported	Comments
400	Bad Request	Yes	The gateway does not generate this response. On reception of this message, before a 200 OK has been received, the gateway responds with an ACK and disconnects the call.
401	Unauthorized	Yes	Authentication support for Basic and Digest. On receiving this message the GW issues a new request according to the scheme received on this response
402	Payment Required	Yes	The gateway does not generate this response. On reception of this message, before a 200 OK has been received, the gateway responds with an ACK and disconnects the call.
403	Forbidden	Yes	The gateway does not generate this response. On reception of this message, before a 200 OK has been received, the gateway responds with an ACK and disconnects the call.
404	Not Found	Yes	The SIP gateway generates this response if it is unable to locate the callee. On receiving this response, the gateway notifies the User with a Reorder Tone.
405	Method Not Allowed	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
406	Not Acceptable	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
407	Proxy Authentication Required	Yes	Authentication support for Basic and Digest. On receiving this message the GW issues a new request according to the scheme received on this response.
408	Request Timeout	Yes	The gateway generates this response if the no-answer timer expires. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.

Table 2-8: 4xx SIP Responses (continues on pages 14 to 15)

4xx Response		Supported	Comments
409	Conflict	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
410	Gone	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
411	Length Required	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
413	Request Entity Too Large	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
414	Request-URL Too Long	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
415	Unsupported Media	Yes	If the gateway receives a 415 Unsupported Media response, it notifies the User with a Reorder Tone. The gateway generates this response in case of SDP mismatch.
420	Bad Extension	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
480	Temporarily Unavailable	Yes	If the gateway receives a 480 Temporarily Unavailable response, it notifies the User with a Reorder Tone. This response is issued if there is no response from remote.
481	Call Leg/Transaction Does Not Exist	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
482	Loop Detected	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
483	Too Many Hops	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
484	Address Incomplete	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
485	Ambiguous	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
486	Busy Here	Yes	The SIP gateway generates this response if the called party is off hook and the call cannot be presented as a call waiting call. On receiving this response, the gateway notifies the User and generates a busy tone.
487	Request Canceled	Yes	This response indicates that the initial request is terminated with a BYE or CANCEL request.
488	Not Acceptable	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.

2.2.5.5 5xx Response – Server Failure Responses

Table 2-9: 5xx SIP Responses

5xx Response		Comments
500	Internal Server Error	On reception of any of these Responses, the gateway releases the call, sending appropriate release cause to PSTN side. The gateway generates 5xx response according to PSTN release cause coming from PSTN.
501	Not Implemented	
502	Bad gateway	
503	Service Unavailable	
504	Gateway Timeout	
505	Version Not Supported	

2.2.5.6 6xx Response – Global Responses

Table 2-10: 6xx SIP Responses

6xx Response		Comments
600	Busy Everywhere	On reception of any of these Responses, the gateway releases the call, sending appropriate release cause to PSTN side.
603	Decline	
604	Does Not Exist Anywhere	
606	Not Acceptable	

3 Known Constraints

3.1 Hardware Constraints

None.

3.2 SIP Constraints

1. SIP transport over TCP is not possible.
2. SDP hold: a=inactive does not put a call on hold. Use a=sendonly or ip=0.0.0.0 to signal hold.
3. Authorization: The gateway supports authorization only for INVITE and REGISTER transactions.
4. rPort deactivation is not supported.
5. DTMF: if telephony events according to RFC 2833 are used, the gateway plays the DTMF tone for the duration specified in the 1st RFC2833 packet. All subsequent packets for the same tone are ignored.

3.3 ISDN Constraints

None.

3.4 SIP / ISDN Gateway Constraints

None.

3.5 IP Constraints

1. Access Control List: CoS 'User Defined' is not mapped to IP packets through the WAN Out Interface.
2. Access Control List: changing an access control rule requires a reboot.

3.6 Web Management Constraints

1. Fallback to factory configuration is not possible through the Web interface. The user must use the CLI or the Reset button.

3.7 CLI Management Constraints

None.

3.8 Maintenance Constraints

1. After a software (SW) downgrade from SW version 2.0.7.5476 to an older version, it's recommended to reset the device to factory defaults. This is due to the replaced CLI prefixes: 'pstn' with 'isdn' for port and interface; 'bluebox' with 'manipulation' for routing configuration. SW versions older than 2.0.7.5476 don't start up with startup configurations generated by SW version 2.0.7.5476. However, the SW version 2.0.7.5476 does startup correctly with older configuration files.

Reader's Notes

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SIP | **MediaPack™ MP-40x**

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